Performance Comparison of Local Area Video Streaming Systems

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***Abstract*—The performance of our Quality Oriented Adaptation Scheme (QOAS) for multimedia streaming in local networks is compared with other existing solutions (TFRCP, LDA+, and non-adaptive). This comparison is done in terms of bandwidth utilization, number of concurrent clients, loss rate, and end-user perceived quality. Simulation results show that for the same average end-user quality, our QOAS system can accommodate a significantly higher number of simultaneous clients while also having higher bandwidth utilization. For the same number of clients, the average end-user quality is always higher for QOAS than for the other solutions studied.**

***Index Terms*—Adaptive video streaming, feedback control, grading scheme, end-user quality.**

I. INTRODUCTION

Currently there is a trend in multimedia presentation towards on-demand-based access to rich media and very high quality multimedia to home residences via an all-IP infrastructure [1, 2]. Network operators and service providers aim for high infrastructure utilization and a large number of customers to increase their revenues. At the same time, customers are interested in receiving high quality streamed multimedia, having access to diverse services, and paying a low cost. This paper analyses how a new Quality Oriented Adaptation Scheme (QOAS) balances these opposing goals in comparison to some well-known streaming approaches like TFRCP [3], LDA+ [4], or a non-adaptive approach.

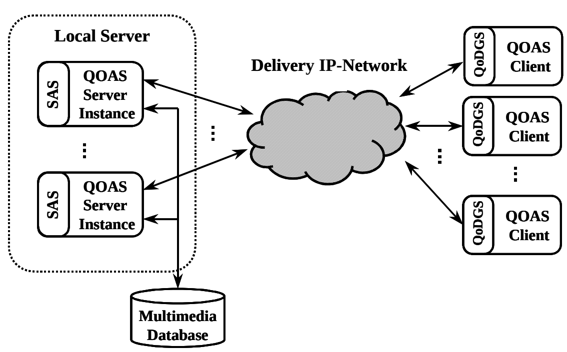

Fig. 1. The QOAS-based multimedia-system architecture

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Different solutions have been proposed to ensure quality of service while streaming multimedia over IP networks [5]. Many of these solutions rely on adaptive streaming schemes that can be classified according to where the adaptation takes place: sender-driven solutions such as TFRCP [3], LDA+ [4], RAP [6] or LQA [7], receiver-driven solutions such as RLM [8] or RLC [9], or transcoding-based solutions (e.g. filters [10]). However none of these solutions take into account end-user perceived quality directly in their adjustment policy. In contrast, our proposed QOAS includes an estimation of end-user perceived quality in its adaptation process, enabling it to achieve higher overall performance.

II. QUALITY ORIENTED ADAPTATION SCHEME (QOAS) FOR MULTIMEDIA STREAMING

The QOAS-based **system architecture** (Fig. 1) includes multiple instances of QOAS adaptive client and QOAS server applications that bi-directionally exchange video data and control packets through the delivery network [11]. The client monitors some transmission and end-user quality-related parameters, and its Quality of Delivery *Grading Scheme* (QoDGS) regularly computes scores that reflect the overall quality of the streaming process. These grades are sent as feedback to the server, whose Server *Arbitration Scheme* (SAS) analyses them and proposes adjustment decisions in order to increase the end-user perceived quality in the reported conditions. Since we limit attention to Video on Demand, each streaming process involves ***one*** server and ***one*** client instance.

For each video streaming process, QOAS defines a number of different **server states** (e.g. a five-state model was used for our experimental tests). Each server state is then assigned to a different stream quality. The stream quality versions differ in terms of compression-related parameters (e.g. resolution, frame rate, colour depth) and therefore have different bandwidth requirements. They also differ in end-user perceived quality. During transmission the server dynamically varies its state according to the reported end-user stream quality. For example, when the client reports a decrease in end-user quality, the server switches to a lower quality state, which reduces the quantity of data sent. In improved viewing conditions, the server gradually increases the quality of the delivered stream.

The **Quality of Delivery Grading Scheme** (QoDGS) [12] at the client (Fig. 2) monitors and evaluates the effect of the delivery conditions on end-user perceived quality. Its grading is based on monitoring both short-term and long-term variations of packet loss rate, delay, and delay jitter, which have been shown to have a significant impact on the received quality [13, 14]. These short-term and long-term periods are

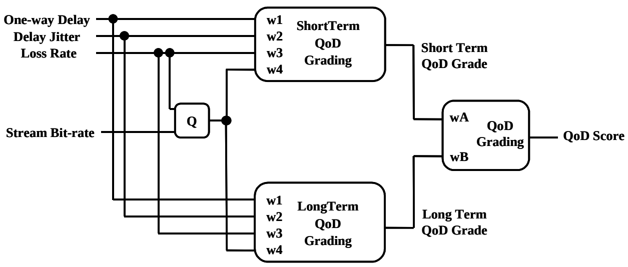

Fig. 2. QoDGS takes into consideration both traffic-related parameters and end-user perceived quality

$$QoD_{ST} = w_1 * DelayG_{ST} + w_2 * JitterG_{ST} + w_3 * LossG_{ST} + w_4 * Q_{ST}$$

$$QoD_{LT} = w_1 * DelayG_{LT} + w_2 * JitterG_{LT} + w_3 * LossG_{LT} + w_4 * Q_{LT}$$

$$QoD_{Score} = w_A * QoD_{ST} + w_B * QoD_{LT} \quad (1)$$

considered, respectively, an order and two orders of magnitude greater than the feedback-reporting interval. The QoDGS also takes into account end-user quality as measured by the multimedia quality metric Q [15], which maps the joint impact of bitrate and data loss on video quality onto the ITU-T R P.910 five-point grading scale [16]. The QoDGS scoring process uses the formulas shown in equation (1). The best results in terms of adaptiveness, link utilization, end-user quality, and stability in local broadband IP networks were obtained experimentally for $w_1 = 0.4$, $w_2 = 0.3$, $w_3 = 0.2$, $w_4 = 0.1$, $w_A = 0.75$ and $w_B = 0.25$.

The **Server Arbitration Scheme (SAS)** considers the values of a number of consecutive QoDGS scores from the client and, by averaging these values, ***asymmetrically*** suggests adjustment decisions. It requires fewer scores to trigger a quality decrease than for a quality increase, ensuring a fast reaction during bad delivery conditions and helping to eliminate its cause. An increase is performed only when the network conditions have improved. This asymmetry helps also to maintain system stability, by reducing the frequency of quality variations.

Since for testing we use the MPEG-2 encoding scheme, our quality adjustment mechanism takes into account the MPEG I-P-B frame-based structure and defines possible adjustment points at the beginning of each Group of Pictures (GOP).

III. EXPERIMENTAL RESULTS

A. *Simulation Models, Setup and Video Sequences*

The experimental tests consisted of simulations using models for QOAS, TFRCP, LDA+ and non-adaptive streaming, built using Network Simulator version 2 (NS-2) [17]. The “Dumbbell” topology used for simulations assumes a single shared bottleneck link with 100 Mbps bandwidth and 100 millisec delay. The other links are over-provisioned so that the only packet drops and significant delays are caused by congestion that occurs on the bottleneck link. The buffering at the bottleneck link uses a drop-tail queue of size proportional to the product of round trip time and bottleneck link bandwidth.

During tests a maximum rate of 4 Mbps was used for Non-Adaptive (NoAd) streaming. The TFRCP implementation used had the update interval M=5 sec as suggested in [3] for delays greater than 100 millisec, as in our setup. The implementation of LDA+ used an RTCP feedback interval of 5 sec as suggested in [4]. Our QOAS model conforms to the description in section II, with a server arbiter upgrade period of 6 sec and a downgrade timeout of 1 sec. The QoDGS short-term period was taken as 1 sec, and the long-term period was 10 sec.

Five five-minute long video sequences were selected from movies with different degrees of motion content: *diehard1* - high, *jurassic3* - average, *dontsayaword* - average/low, *familyman* - low and *roadtoeldorado* (cartoons) - average /high. The clips were MPEG-2 encoded at five different rates between 2 Mbps and 4 Mbps using the same frame rate (25 frames/sec) and the same IBBP frame pattern (9 frames/GOP).

B. *Simulation Scenarios and Results*

The simulations involved a number of clients randomly selecting both the movie clip and the starting sequence from within the chosen clip. The resulting video streaming

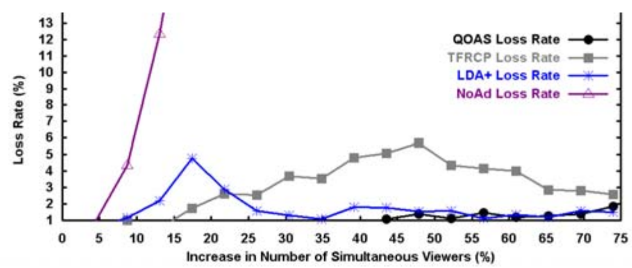

Fig. 3. Loss rate versus increase in the number of clients simultaneously served above a base line of 23

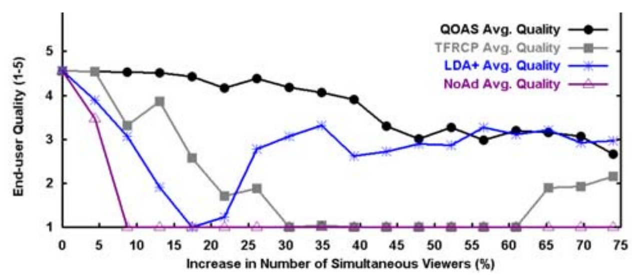

Fig. 4. End-user average quality versus increase in the number of clients simultaneously served above a base line of 23

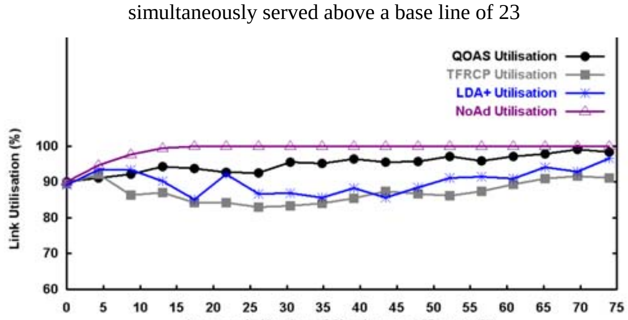

Fig. 5. Bottleneck link utilization using different approaches, while increasing the number of simultaneous viewers

processes began and ended during transitory periods of 50 sec duration, which were not taken into account when analysing the results. The length of the stable periods taken into account in each case was 150 sec.

The QOAS, TFRCP, LDA+ and NoAd approaches were used in turn as the video streaming method, and the number of clients was gradually increased above a base line of 23 in each case. This number of clients was chosen because it allowed for lossless streaming and maximum end-user perceived quality in each of the four cases. Fig. 3 shows the loss rate as a function of the increase in the number of simultaneously served clients, Fig. 4 presents the end-user quality as a function of the increase in the number of simultaneously served clients, and Fig. 5 plots the bottleneck link utilization values when the number of clients similarly increases. The end-user perceived quality was measured by the multimedia quality metric Q [15] on the ITU-T R P.910 five-point grading scale [16].

According to these tests, the number of simultaneous users served is significantly higher for QOAS in comparison with the other streaming schemes. For example, to maintain a “good” perceptual quality level, by using QOAS 23% more clients could be served than by using TFRCP, 33% more clients than by using LDA+, and 39% more users than by using the NoAd solution. If the goal is to maintain a “fair” average quality level for the clients, the benefit of using QOAS is 26% greater than TFRCP, 13% greater than LDA+, and 42% greater than NoAd.

In terms of efficient usage of available bandwidth, QOAS was superior at all times to TFRCP and LDA+-based streaming. Using QOAS, the bottleneck link utilization exceeded 95% for 30 simultaneous clients and reached 99% for 40 clients. The values obtained for TFRCP and LDA+ are more modest: around 84% and respectively 87% for 30 simultaneous clients, and 92% and respectively 96% for 40 clients. Under the same conditions, the 100% figures obtained by NoAd came with severe costs in terms of loss and significantly reduced end-users quality.

QOAS facilitates the choice of network load level according to economic, technical, and quality goals. It seems likely that service operators will maximise their revenues from offering VOD services to an increased number of clients while delivering a target quality level. For example, by scaling the simulation results with the “good” target quality level to a gigabit Ethernet connection, QOAS could service 320 simultaneous users compared to only 260 using TFRCP, 240 using LDA+, and 230 using NoAd streaming.

IV. Conclusions and Further Work

This paper compares our Quality Oriented Adaptation Scheme (QOAS) with three other solutions for video streaming: TFRCP, LDA+, and non-adaptive (NoAd). Simulation results show that for the same average end-user quality, QOAS can accommodate a significantly higher number of simultaneous clients while also having higher bandwidth utilization. For the same number of clients, the average end-user quality is always higher for QOAS than for the other solutions studied.

Further work will test the effect of changes in the feedback interval on the performance of these adaptive mechanisms, and their behavior when other types of traffic (e.g. short-lived TCP, long-lived TCP, non-adaptive UDP) share the same link with these adaptive streams. We also plan to carry out subjective perceptual tests on a prototype system to verify the end-user quality results presented here.

Since we have focused on the performance benefits QOAS provides in terms of end-user perceived quality, loss rate and link utilization, its TCP-friendliness was not addressed. However future work could explore QOAS’s degree of TCP-friendliness and compare it to those of the other schemes. Future work could also address the relationship between the degree of TCP-friendliness and the adaptiveness of these schemes.

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